REMARKS

This Request for Continued Examination is being filed together with an amendment that includes rewritten claim 3, and arguments concerning the rejection of the claims as being obvious in view of U.S. Patent No. 6,519,261 issued to Brueckheimer, et al. ("Brueckheimer"), and U.S. Patent No. 6,683,877 issued to Gibbs, et al. ("Gibbs"). The amendment adds no new matter and is supported in the Specification as filed, for example, at Figs. 1 and 2, as well as in the text, at pages 7-12.

The Final Office Action at page 2 reports independent claims 1, 3, 5, and 11 as being obvious over Brueckheimer and Gibbs. Applicant respectfully disagrees because none of the cited portions of Brueckheimer or Gibbs teach or suggest, for example referring to claim 1, *dynamically establishing AAL2 channel identifiers on a call-by-call basis using ATM standards-based call control signaling protocols and mapping the CIDs to virtual path/virtual channel in a standards-based ATM call control protocol.*

Beginning with Brueckheimer, although Brueckheimer describes the use of AAL2 (and more particularly a inter-working framework between IP, time division multiplex, and ATM networks, Brueckheimer does not teach or suggest the use of an ATM standards-based call control signaling protocol or AAL2 switched voice networking. Taking the citations in sequence given on page 2 of the Final Office Action, column 2, lines 40-50 of Brueckheimer describe AAL2 but do not refer to any call control signaling to establishing paths through an ATM network.

The next citation is column 5, line 67 through column 6, line 10, which refers to CAS/CCS signaling. However, as can be seen in the enclosed printout of the web page www.pulsewan.com, which gives a novice guide to CAS/CCS, neither of those signaling methods are ATM standards-based or ATM Forum promulgated. Both CAS and CCS are used for T1 time division multiplex systems. There is no teaching or suggestion of replacing CAS/CCS with an ATM standards-based signaling protocol in Brueckheimer.

Next, the Final Office Action points to the Abstract of Brueckheimer. There, the inter-working function is described to interface between IP, TDM, and ATM networks. However, there is once again no teaching or suggestion for establishing AAL2 calls using ATM standards-based call control signaling, as no call control signaling is mentioned. The same holds for column 1, lines 4-6 of Brueckheimer which merely refer to telecommunication networks in general, and methods for adapting traffic in such networks between different payload protocols (IP, TDM, and ATM).

The next citation is column 2, lines 40-60 of Brueckheimer, which again only generally refers to AAL2 and its 47-byte payload structure. It is mentioned that telephony channels can be adapted into corresponding mini packets which are multiplexed into a single virtual connection that can be deemed a variable rate pipe. Connections within the pipe can be resized dynamically, started and ended while the virtual connection is continually active. Once again, however, there is no teaching or suggestion or any particular technique for establishing, maintaining and clearing of such AAL2 network connections.

Next, the Final Office Action cites column 3, lines 25-26. This section mentions nothing about call control signaling. Getting back to column 1, lines 50-60, an adaptation layer used for continuous bite rate services is described. However, there is no mention of a standards-based control signaling protocol, CAS/CCS as submitted above is not ATM standards-based nor is it ATM Forum promulgated.

Finally, the Final Office Action at the bottom of page 2 cites column 5, line 65 to column 6, line 9 as also teaching of Applicant's claimed signaling protocol. However, in that section of Brueckheimer, CAS or CCS is described to be used in the inter-working function of Fig. 1, and as mentioned above, CAS and CCS are not ATM standards-based protocols.

Next, at page 3 of the Final Office Action, Gibbs is cited as allegedly teaching "executing a call set up process in the AAL2 signaling layer", at column 5, lines 45-59 and 29-42. However, the closest Gibbs appears to come to teaching any form of *call control signaling to establish AAL2 CIDs on a call-by-call basis* is a method of setting up a voice connection in a multimedia session between first and second gateways by

exchanging session descriptors between the gateways and providing in the session descriptors information whereby the gateways negotiate and agree on adaptation of attributions for the voice connection. [Gibbs, column 2, lines 36-65] Gibbs later refers to a **SDP session descriptor** that incorporates the identity of a virtual channel connection to establish the virtual channel connection between the gateways. [Gibbs, column 4, lines 44-53] However, SDP does not teach or suggest *using ATM standards-based call control signaling*.

In sum, although Brueckheimer discusses a VC being deemed a variable rate pipe that can be resized dynamically, there is no teaching or suggestion of using ATM standards-based call control signaling or ATM Forum promulgated signaling protocols for establishing on a call-by-call basis AAL2 switched voice connections on an end-to-end basis. It should be noted that both ends of the call need not be AAL2 switched, but rather at least one end has an AAL2 switched voice connection that is dynamically established on a call-by-call basis. Neither the CAS/CCS discussed in Brueckheimer nor the SDP of Gibbs teaches or suggests, for example, *mapping AAL2 channel identifiers to a virtual path/virtual channel within a standards-based ATM call control protocol*. For the above reasons, the art rejection of the claims is improper and should be withdrawn.

If necessary, the Commissioner is hereby authorized in this, concurrent and future replies, to charge payment or credit any overpayment to Deposit Account No.

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02-2666 for any additional fees required under 37 C.F.R. §§ 1.16 or 1.17, particularly, extension of time fees.

Respectfully submitted,

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Dated: October 31, 2005

By ______________________
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I hereby certify that this correspondence is being deposited with the United States Postal Service as first class mail with sufficient postage in an envelope addressed to: Mail Stop RCE, Commissioner for Patents, Post Office Box 1450, Alexandria, Virginia 22313-1450 on October 31, 2005.

Margaux Rodriguez *October 31, 2005*

What is CAS/CCS and R2 ?

Channel Associated Signaling
Common Channel Signaling

PDF Available

Presented by: aculab

CAS - *A Novice Guide*

An expanded version of this article is available on the PDF file

Introduction

CAS stands for **Channel Associated Signalling**. With this method of signalling each traffic channel has a dedicated signalling channel. In other words the signalling for a particular traffic circuit is permanently associated with that circuit. This makes CAS inflexible and slow.

Channel-associated call-control is still widely used today mostly in South America, Africa, Australia and in Europe. However since 1979 other forms and applications of signalling have come about and can be generally referred to as Common-Channel signalling (CCS).

Common Channel Signalling (CCS) was introduced in 1976 and is where the common channel carries data messages which convey signalling for the circuits between two switches. CCS only requires one signalling channel for up to 1000 traffic channels. It is able to do this by only signalling when required, unlike CAS which signals even if nothing has happened. CCS is faster, more flexible and allow greater services

T1 Bearer

T1 describes a multi-channel system used in Northern America and Japan. This combines 24 input channels sampled at 8kHz, each carrying an 8 bit digital word, using mu-law encoding (similar to A-law used in Europe). An additional frame alignment bit is added per frame giving 1.544Mbps aggregate (the sum of the 24 channels) signal.

Line Signalling

The 4 bits available in timeslot 16 for signaling allows for 16 possible signalling states, this much are seldom used or required. These signals are know as line signalling, supervision signals or ABDC bits. These signals are used to set up and clear down the call and represent events that occur on the trunk such as seizure, proceed-to-send, answer, clear forward, etc. While the majority of supervision signals are used in all CAS systems, there are system-specific differences in the sets of supervision signals.

In **E&M** (Ear and Mouth) line signalling the speech circuit (or channel) has an associated E-wire and M-wire for signalling. In this type of line signalling only one bit of the signalling changes at any one time.

Register signalling

Register signalling also known as Address Signalling, selection signals and digits. The digits are used primarily to indicate the called number, but can also have other meanings. Examples of register signalling are DTMF, MFC R2, decadic (Loop disconnect) and MFR1.

Loop disconnect Signalling (or decadic) is associated with a DC analogue CAS system and was used in the early CAS systems. In this situation the local switch provides a DC voltage on all subscriber lines enough to power a telephone. When the telephone is on-hook (idle), the loop inside the instrument is broken, and no lines current is drawn. When the subscriber goes off hook, initiating a call, current is drawn. The sending of the dial digits causes the loop to be opened and closed at a rate of 10 pulses per seconds. Thus each number in the dial (0 to 9) can be represented by a series of pules and the digit 0 to equal to 10 pulses. The decadic pulsing can be seen via the line signalling by the toggling of one of the ABCD bits (usually the A-bit).

The two main disadvantages of Loop disconnect (LD) are; slow signalling speed and the requirement for a metallic path. Allowing for interdigit pauses LD signalling can transfer approximately 1 digit per second. LD is not suited to carried systems (FDM) or radio systems due to its' need for a metallic connect between the subscriber and the switch.

MF (Multi-frequency) signalling uses a two-tone combination to represent a dialed digit and is usually associated with push button phones. The tones are chosen from within the voice band (in-band) and transmitted as audio tones over the traffic circuit. A single tone is considered unsuitable due to possible voice imitations.

MF is much faster than LD as it is capable of transferring several digits per second. The ITU-T standard MF system is number 4, (MF4). This signalling technique is also referred to as Dual Tone Multi-frequency (DTMF).

One of the more familiar CAS protocols is **MFC R2**. This is a compelled sequence multi-frequency code signalling. The fundamental principles of compelled multi-frequency code register signalling were developed in 1954. In 1968 this signalling systems was recognized by CCITT as an international signalling system for regional use. MFC R2 can be used on international as well as national connections.

In MFC R2 signalling, the equipment units at the exchanges that send and received digits, and the signalling between these units, are usually referred to as register and interregister signalling.

The **compelled signalling operates as follows**:

- On seizure of a link (or line), the outgoing R2 register automatically starts sending the first forward interregister signal;
- as soon as the incoming R2 register recognizes this signal, it starts sending a backward interregister signal which has it's own meaning and at the same time serves as an acknowledgement signal;
- as soon as the outgoing R2 register recognizes the acknowledging signal, it stops sending the forward interregister signal.
- as soon as the incoming R2 register recognizes the cessation of the forward interregister signal, it stops sending the backward interregister signal;

- as soon as the outgoing R2 register recognizes the cessation of the acknowledging backward interregister signal it may, if necessary, start sending the appropriate next forward interregister signal.

Back Busy

Back busy (or blocking) is a signal that is available in some CAS (E1 and T1) protocols, that is sent over the ABCD bits, and is interpreted at the far end that the channel is not available for call placement (incoming to this end).

There is sometimes the concept of one-way working and both-way working lines, and back busy is typically only available against the direction of the call (in the backward direction). Back busy may even be illegal in the forward direction of lines that are configured for one-way use, but which are otherwise capable of both-way working (e.g. R2).

Without the 'auto back busy' functionality, when a channel is released to allow it to be used for another call, it returns immediately to the idle state, even though the application that would own it has not yet prepared to accept another call on that channel. Thus, it is possible for the channel to receive an incoming call before the application is ready to receive and process an incoming call.

With the 'auto back-busy' functionality activated, when a line is released to be used for another call it goes first into the back-busy state, which is interpreted by the far end as unavailable. When an application opens that channel (thus waiting for a call on that channel), the back-busy signal is removed, making the channel once again ready to accept a call.

In the case of R2 can exchange can block an idle trunk by changing its status from a,b = 1,0 to a, b = 1,1. As mentioned before exchanges will not seize these trunks. To end blocking, the exchange returns to a,b = 1,0 (idle).

Terms and Acronyms

DDI Direct dial in

An **outgoing trunk** seizes an outgoing line, sends forward signals, and receives backward signals. An **Incoming trunk** receives forward signals, and sends backward signals.

In-band tones are audible tones, between 300 and 3400Hz.

Out-of-band tones. This is a narrow band of tones used as signalling tones centered at the signalling frequency f=3825 Hz.

ANI. The sending of the calling numbers is known as Automatic Number Identification.

The **Originating exchange** in a call is the local exchange serving the calling subscriber, and the **terminating (or destination) exchange** is the local exchange of the called subscriber.

Overlap address signalling. This is when the called number is not received all at once, so the called digits will be forwarded on to another exchange one at a time.

En-bloc address signalling. This is when the complete called number is sent out in one uninterrupted stream. ISDN protocols usually send their digits in this way.

Link-by-link signalling. This is signalling by two exchanges at the two ends of a trunk.

End-to-end signalling. In the end-to-end address signalling, the digit sender in the originating exchange sends address signals successively to digit receivers in the second, and later exchanges in the connection.

Malicious call holding is another name for last party release

A **Time Slot** is the same as a channel. A timeslot consists of 8 bits containing PCM encoded speech.

A **Frame** consists of all 30 timeslots (in the case of E1, 24 in the case of T1). Each frame contains a sample from each timeslot.

A **Multiframe.** It is not possible for all 30 channels to signal within the 8 bits in time slot 16. Therefore channels take turns using slot 16. Two channels send their ABCD signaling bits in each frame. The 340-user channel then takes 15 frames to cycle through all the signalling bits. One additional frame is needed to synchronize the received to the signalling channel. So the full multiframe has 16 frames.

Tone and Announcements. These include ring tones, busy tones, etc.

A **Register.** In R2 signalling, the equipment units at the exchanges that send and receive digits, and the signalling between these units, are usually referred to as register and interregister signalling.

Meter; Metering signals are pulsed type signals transmitted backwards during the conversation from the call charging point to the subscriber's call meter in the originating exchange. The are used to advise the originating exchange of the estimated cost for a particular dialed call.

Reference:

Wray Castle course notes Signalling Systems in Modern Telecoms Networks.

130 4 13 Ue March 1974, compelled sequence multi-frequency code signalling, Telefonaktiebolaget LM Ericssson, Telephone Exchange Division, S-126 25 Stockholm, Sweden.

Blue book Recommendation Q.440, Q.441 Q.421 Q.422

The Blue book produced by ITU (International Telecommunication Union) and CCITT (The International Telegraph and telephone Consultative Committee) Volume VI – Fascicle V1.4 Specification of signalling systems R1 and R2 Recommendations Q.310 – Q.490.

Signalling Telecommunication Networks by John G. Van Bosse. Published by Wiley –Interscience. ISBN number 0-471-57377-9